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TITLE

VOICE OVER INTERNET COMMUNICATIONS ALGORITHM AND RELATED METHOD FOR OPTIMIZING AND REDUCING LATENCY DELAYS

FIELD OF THE INVENTION

The present invention relates to a communications algorithm and related method for optimizing and reducing latency delays in voice over Internet Protocol transmissions.

CROSS REFERENCE TO RELATED APPLICATION

This application claims the benefit of U.S. Provisional Application No. 60/255,582, filed December 14, 2000, the contents of which are incorporated herein by reference in their entirety.

BACKGROUND AND SUMMARY OF THE INVENTION

Internet Protocol ("IP"), or Voice over IP ("VoIP") enables the transmission of voice calls and other multimedia over data networks such as the Internet or corporate enterprise networks. Local IP Telephony gateways act as interfaces between local telephone and data networks to pick-up and hand-off calls. At the gateways, signals are compressed and decompressed, packetized and reassembled. Unfortunately, with this emerging technology, significant problems remain in connection with data packet loss and latency delays during the transmission of data packets. When these data packets embody voice calls, packet loss and latency delays can destroy voice recognition and the quality of the desired communication. Such latency delays arise, in part, from the different modes of data transmission, e.g., fiber optics, hard wiring, satellite communications, cellular transmissions, etc., and the related band width restrictions associated with such transmissions.

Dedicated communications lines and networks (such as a fiber optic cable), with controlled and select data transmission, can obviate latency delays. However, the cost of constructing such a network between individual gateways can be prohibitive.

Thus, a need exists for a VoIP communications algorithm and related method which optimizes latency delays in the generally available IP network or in the "public" Internet. As

set forth more fully below, a solution to latency delays during VoIP communicationscan be obtained through use of IP version 6 ("IPv6") and the introduction of a new header in IPv6 for latency or delay which interacts with a dynamic library to continually test and select the best route of transmission through the IP network and the "public" Internet. This novel VoIP communications algorithm and related method is scalable and affords opportunities for reduced telephony fees and unlimited call potential with reduced latency and improved quality of sound.

Other objects and advantages of the present invention will become readily apparent to those skilled in this art from the following detailed description. The embodiments shown and described provide illustration of the best mode contemplated for carrying out the invention. The invention is capable of modifications in various obvious respects, all without departing from the invention. Accordingly, the drawings are to be regarded as illustrative in nature, and not as restrictive.

BRIEF DESCRIPTION OF THE DRAWINGS

FIG. 1 is a block diagram of the VoIP communications algorithm and related method of the present invention.

FIG. 2 is a block diagram of an internet system embodying the VoIP communications algorithm and related method of the present invention.

DETAILED DESCRIPTION OF THE PREFERRED EMBODIMENTS

The present invention provides a VoIP communications algorithm and related method which minimizes latency delays associated with data packet transmission. In particular, the present invention employs IPv6 in conjunction with latency or delay header and a dynamic library which continually tests and selects the best transmission route through the "public" Internet. This novel VoIP communications algorithm and related method is scalable and affords opportunities for reduced telephony fees and unlimited call potential with reduced latency and improved quality of sound. Other multimedia applications are also possible.

The Internet Protocol or IP has its roots in earlyresearch networks of the 1970s, but

within the past decade has become the leading network layer protocol. This means that IP is a primary vehicle for a vast array of client server and peer to peer communications, and the current scale of deployment is straining many aspects of its twenty-year old design. Further, efforts to successfully implement VoIP technology has encountered problems arising from data packet loss and latency delays associated with transmission of voice signals, as data packets, through the network. In particular, variable delays in transmission can destroy the recognition of speech. Many of these problems arise from limitations inherent in the current IP version in use – IP version 4.

The present invention implements the benefits which are provided through IPv6. More specifically, IPv6 has been designed to enable high performance, scalable internetworks that should operate as needed for decades. IPv6 offers a number of enhanced features, such as a larger address space and improved packet formats. IPv6 also regularizes and enhances the basic header layout of the IP packet. In this new header format, concepts of main and additional, optional headers are introduced. The main header format in IPv6 is as follows:

Version = 6	8 bits Traffic Class	20 bits Flow Label
16 bits Payload Length	8 bits Next Header	8 bits Hop Limit
128 bits Source Address		
128 bits Destination Address		

In this header format, the protocol type field of IPv4 (e.g., the TCP or User Datagram Protocol (UDP)) has been replaced by the "Next Header" field. Each header field indicates the type of the next header, which can be a TCP/UDP header, or another IPv6 header. Extension headers for IPv6, in the suggested order, are as follows:

(Primary IPv6 NEXT header)

Hop-by-Hop Options header

Destination Options header – 1

Source Routing header

Fragmentation header

Authentication header

IPv6 Encryption header

Destination Options header– 2

These extension headers are followed by upper layer headers and payload. Each extension header typically occurs only once within a given packets, with the exception of the destinations option header.

In the present invention, an additional header for Latencyor Delay is added to these extensions of the NEXT header format. In particular, the present invention involves a modification of the Source Routing headerwithin the NEXT header extension format.

As detailed below, this additional Latency header, as a modification of the Source Routing header, works in correlation with the existing extension headers within IPv6 to measure and analyze the latency or delay on all available routes through the "public" Internet and select the optimal route at a given period of time and for a given destination.

More specifically, within IPv6 (as in IPv4) a signal/feedback or trace program exists which allows an IPv6 to test route options for a data packet transmission. The VoIP communications algorithm and related method of the present invention augments this trace program and the Hop by Hop Options header of IPv6 to optimize route transmission and reduce latency on a consistent and regularized basis. In operation, the VoIP communications algorithm of the present invention includes a dynamic data library, otherwise known as "DLL," which contains both a destination database and a routing database, which optimizes route transmission in correlation with the Hop-by-Hop Options header. When activated by a new data packet transmission, IPv6 will initially run its trace program to test for the best transmission route. The VoIP communications algorithm of the present invention adds a gross latency database, an optimizations block, a criterion block and a quality of routes analyzer which, in conjunction with the dynamic data library or DLL and IPv6, constantly

retest for the best transmission route based on both current and past network conditions for a given destination at a given route.

Referring to FIG. 1, block diagram **10** depicts the manner in which the novel VoIP communications system and method of the present invention interacts with IPv6. In diagram **10**, router **11** and gateway **12** are standard hardware devices which are integrated with IPv6. Router **11** and gateway communicate to both the PSTN and the internet. Within IPv6, next headers generator **13**, hop by hop options generator **14**, current route generator **15** and router manager **16** are standard components.

The VoIP communications algorithm and related method of the present invention adds to IPv6 the following components and data packet processing steps: dynamic data library or DLL **17** (which includes destination database **18** and routing database **19**), gross latency database **20**, optimizations block **21**, criterion block **22**, quality of routes analyzer **23** and hop by hop switch **24** which is based on and works in correlation with DLL**17**.

In operation, a data packet received through gateway **12** will trigger the trace program of IPv6, whereby all available routes through the "public" Internet will be identified, along with the time or latency associated with each route. The communications algorithm and related method of the present invention, as a next step, measures and identifies a subset of all available routes which represent the best available routes based upon latency and possible other factors, such as data packet volume. More specifically, gross latency database **20** and quality of routes analyzer **23** utilize preset criteria to measure and create a subset of best available routes for a given data packet transmission. These criteria are set by criterion block **22**, are applied by gross latency database **20** and the results are measured and analyzed by quality of routes analyzer **23**. The resulting "best available routes" subset is created a database within routing database **19**. For example, criterion block can establish criteria for "best available routes" based upon latency measured at less than 100 ms. The criterion block also establishes the number of best available routes, e.g., 5, 10, 20, etc., which will be created as a database within routing database **19**. To the extent data packet volume is also to be

tested, criterion block **22** would be preset with relevant test factors in this respect as well.

Utilizing the same criteria, i.e., latency and other such data packet volume, the communications system and method of the present invention next identifies the best route, i.e., the route with the least latency, for the data package at that given period of time. In operations, this measurement is also made by the quality of routes analyzer and the resulting information is stored within routing database **19**.

As a next step, the novel VoIP communication algorithm of the present invention selects the optimal route, with the least latency, at the precise transmission time of the data packet based on both current and historical information as the best route. In operation, this selection process involves optimizations block **21**, destination database **17** and routing database **18**. More specifically, destination database **17** and routing database **18** contain a routing table or historical information concerning the optimal transmission route for a data packet being directed to a specific destination at a specific time. For example, for a transmission between identified addresses in the United States and South America, destination database **17** and routing database **18** contain historical information showing the best route, i.e., the route with the least latency, at a given time within a day, week and year. It may be that the best route at a given instant is anomalous and that historical information within the routing table of destination database **17** and routing database **18** show one of the other routes within the subset of best available routes to be optimal. The route optimization step therefore may override the selection of the shortest available route at a given period of time. This optimal route selection is also based upon criteria set by criterion block **22**. In operation, this optimal route is selected through the IPv6 components of the hop-by-hop options generator **14**, current route generator **15** and router manager **16**.

The communications algorithm of the present invention next continues to use the route management abilities within IPv6 to rerun the trace program at preset, periodic intervals and, through the steps described above, and identify subsequent (i) best available routes subsets, (ii) best routes at a given transmission time and (iii) optimal routes. Trace program intervals,

e.g., every five seconds, are set by criterion block**22**.

Utilizing optimizations block **21** and the hop-by-hop options generator **14** within IPv6, the VoIP communications algorithm and related method of the present invention at this point provides for the option of selecting an alternative best route and switching to same based upon both current and historical route data. In this context, switch **24** prevents hop-by-hop options generator **14** from switching to the best route, based solely on current route data, at each trace program interval and corresponding best route analysis. This switch is turned on by dynamic data library **17**, and the hop-by-hop options generator is guided by routing database 18, when a new optimal route is identified. It is significant, in the present invention, that the dynamic data library **17** and optimizations block **21** bring historical routing table information into this route switching analysis. More specifically, it is not desirable to continue switching from best route to best route, based on instantaneous and periodic latency measurements along, as this can lead to distortions in voice quality. The communications method of the present invention functions to normalize latency delays, in an optimum range, over time so that latency is nominal and voice quality is maintained. In this context, the optimizations block **21** may override a best route, based on instantaneous, current trace measurements for, e.g., five trace intervals. At this point, historical table routing data may be over-ridden in favor of current latency measurements.

As a final step, the communications algorithm and related method of the present invention periodically updates the historic routing table information in dynamic data library **17** based upon measurements of the best available routes subsets and corresponding creations of datasets within routing database **19**. Criterion block **22** sets these update intervals, and quality of routes analyzer acts to delete the accumulated best available routes datasets after this update process is complete. Thereafter, a new cycle of creating best available routes datasets begins anew. Accordingly, the communications system and method of the present invention has the ability to teach itself about changing route and latency conditions.

In operation, the present invention is functionalonly if it is installed in the gateways and POPs at both the transmitting and receiving ends of a voice transmission. Referring to FIG. 2, the VoIP communications algorithm of the present invention, depicted as latency optimizer **30,** is installed at the gateway of each participant in the communications system. These gateways must be IPv6-integrated as well. Once installed, two-way VoIP communications, originating from either participant, are possible between enabled gateways and through routers **38**. Importantly, each individual participant need not have his or her latency optimizer **30**; rather, the gateway and related POP serve as a clearinghouse for multiple users. Thus, in FIG. 2, latency optimizer **30** associated with gateway **31** in the USA/Canada can support multiple communications from difference users, including users with a cell phone **32**, and desk phone **33** or a personal computer **34**. The related PSTN 35 in the USA/Canada can also be accessed. Again, provided the gateway **35** in, for example, the Ukraine is enabled with IPv6 and latency optimizer **30**, users in the USA/Canada can communicate with the multiple users in the Ukraine who are utilizing similar devices, i.e., cell phone **36** or desk phone **37**, and communicating through PSTN**39**.

Described above has been an algorithm and related method for optimizing latency delays in VoIP and other multimedia communications over the "public" Internet. In this disclosure, there is shown and described only certain preferred embodiments of the invention, but, as aforementioned, it is to be understood that the invention is capable of use in various other combinations and environments and is capable of changes or modifications within the scope of the inventive concept as expressed herein.